# Configuration et dépannage de l'enregistrement des appels de base

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## Introduction

Ce document décrit les bases de l'enregistrement des appels dans Cisco Unified Communications Manager (CUCM).

## Conditions préalables

### Exigences

Cisco recommande que vous ayez connaissance de CUCM intégré à un serveur d'enregistrement tiers.

### Composants utilisés

Les informations contenues dans ce document sont basées sur les versions de matériel et de logiciel suivantes :

- CUCM
- Protocole Internet (IP) Cisco
- Serveur d'enregistrement des appels téléphoniques

The information in this document was created from the devices in a specific lab environment. All of the devices used in this document started with a cleared (default) configuration. Si votre réseau est en ligne, assurez-vous de bien comprendre l’incidence possible des commandes.

## Informations générales

Ce document traite également du flux multimédia attendu, des flux d'appels attendus pour les périphériques SIP (Session Initiation Protocol) et SCCP (Skinny Client Control Protocol), ainsi qu'un exemple d'échec courant de la configuration de l'enregistrement des appels.

## Types d'enregistrement des appels

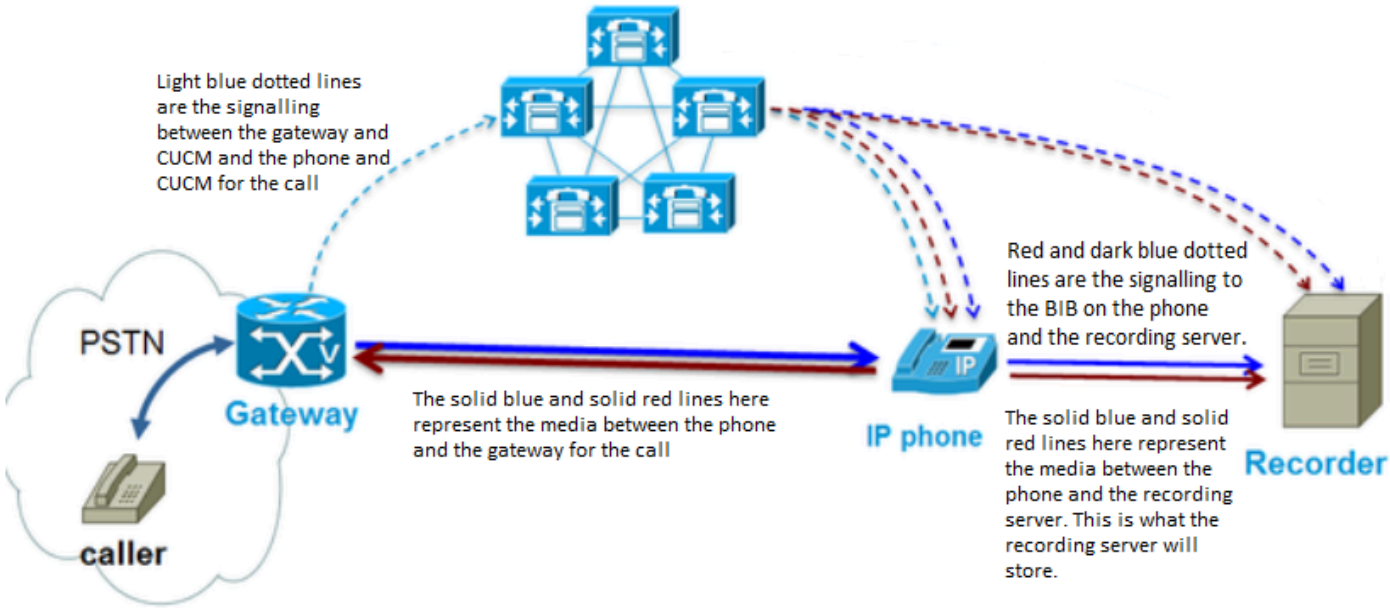
### Automatique

Les principaux éléments de l'enregistrement automatique des appels sont les suivants :

- Utilise le BIB (Integrated-In-Bridge) du téléphone IP pour acheminer l'audio vers la destination d'enregistrement.
- Lancé chaque fois que le téléphone IP passe un appel ou reçoit un appel.
- Nécessite uniquement une liaison SIP entre CUCM et la destination d'enregistrement. Certains fournisseurs d'enregistrement requièrent l'intégration de la téléphonie à l'ordinateur (CTI).
- N'autorise pas l'enregistrement des téléphones situés en dehors du réseau géré (doit avoir accès pour envoyer RTP directement au serveur d'enregistrement et être un téléphone IP

Cisco capable d'allouer un BIB).

Dans ce schéma, les traits pleins représentent le flux de média attendu et les traits pointillés le flux de signalisation attendu :

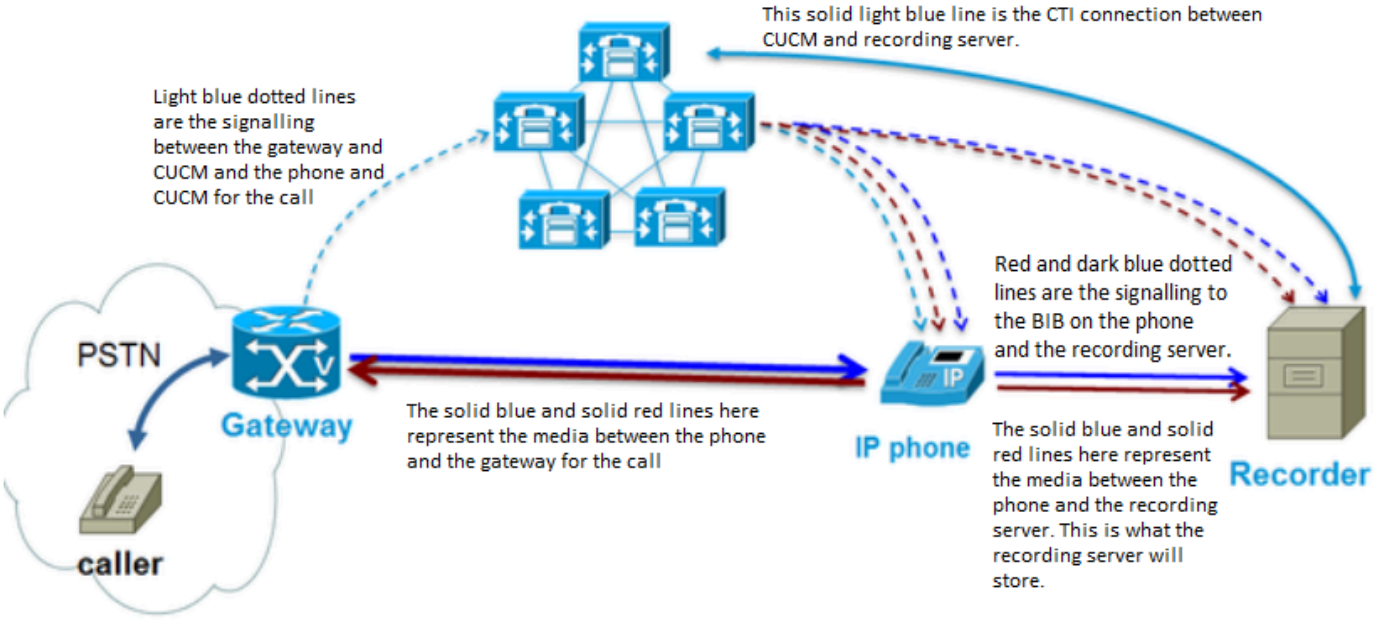



## Application appelée

Les éléments clés de l'enregistrement des appels appelés par l'application sont les suivants :

- Utilise le BIB du téléphone IP afin de transmettre l'audio à la destination de l'enregistrement.
- Lancé lorsque l'application (enregistreur) l'exige.
- Nécessite une ligne principale SIP et CTI avec application d'enregistrement.
- L'utilisateur de l'application CTI doit avoir accès aux terminaux qui doivent être enregistrés.
- N'autorise pas l'enregistrement des téléphones situés en dehors du réseau géré (doit avoir accès pour envoyer RTP directement au serveur d'enregistrement).

Dans le schéma, les traits pleins représentent le flux de média attendu et les traits pointillés le flux de signalisation attendu. La ligne continue entre CUCM et le serveur d'enregistrement indique une connexion CTI entre CUCM et l'application.

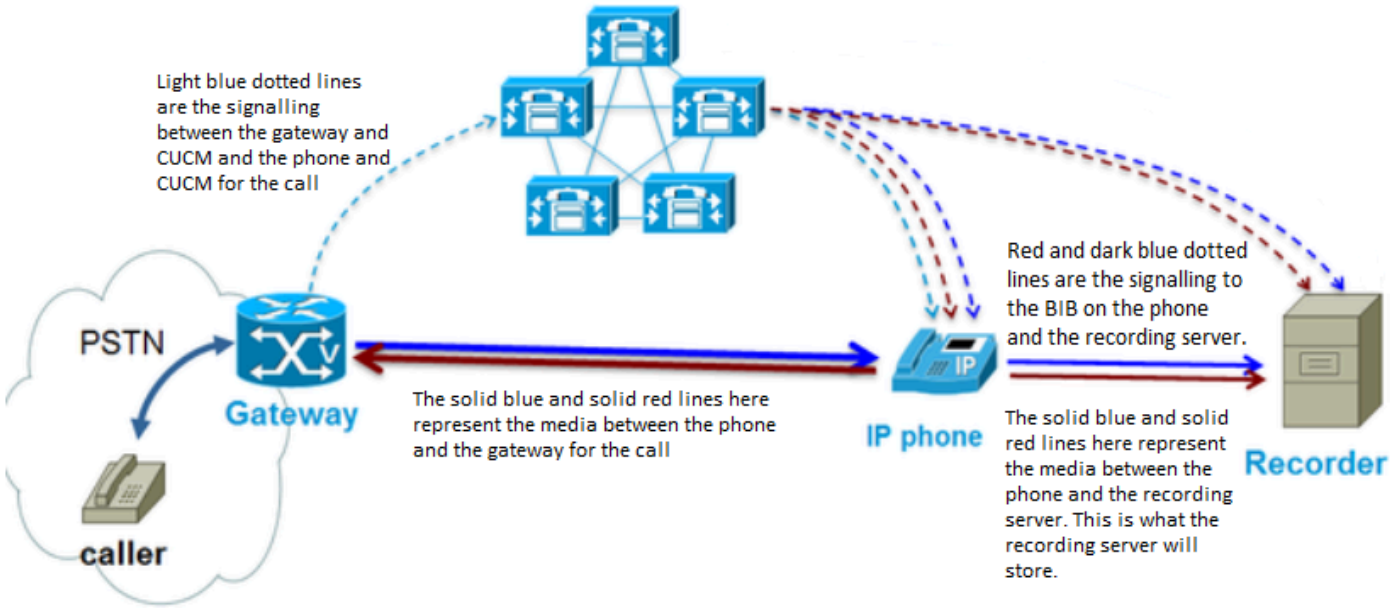



## Sélectif

Les éléments clés de l'enregistrement sélectif des appels sont les suivants :

- Utilise le BIB du téléphone IP afin de transmettre l'audio à la destination de l'enregistrement.
- Lancé à chaque fois que l'utilisateur du téléphone IP sélectionne l'option d'enregistrement sur son téléphone IP (CUCM 9.x+) ou sur une application telle que dans cette image
- Nécessite généralement uniquement une ligne principale SIP entre CUCM et la destination d'enregistrement (qui dépend du fournisseur de l'application d'enregistrement).
- N'autorise pas l'enregistrement des téléphones qui se trouvent en dehors du réseau géré (doit avoir accès pour envoyer RTP directement au serveur d'enregistrement).

Comme vous pouvez le voir dans ce diagramme, le chemin de média et de signalisation est très similaire à l'enregistrement automatique des appels :

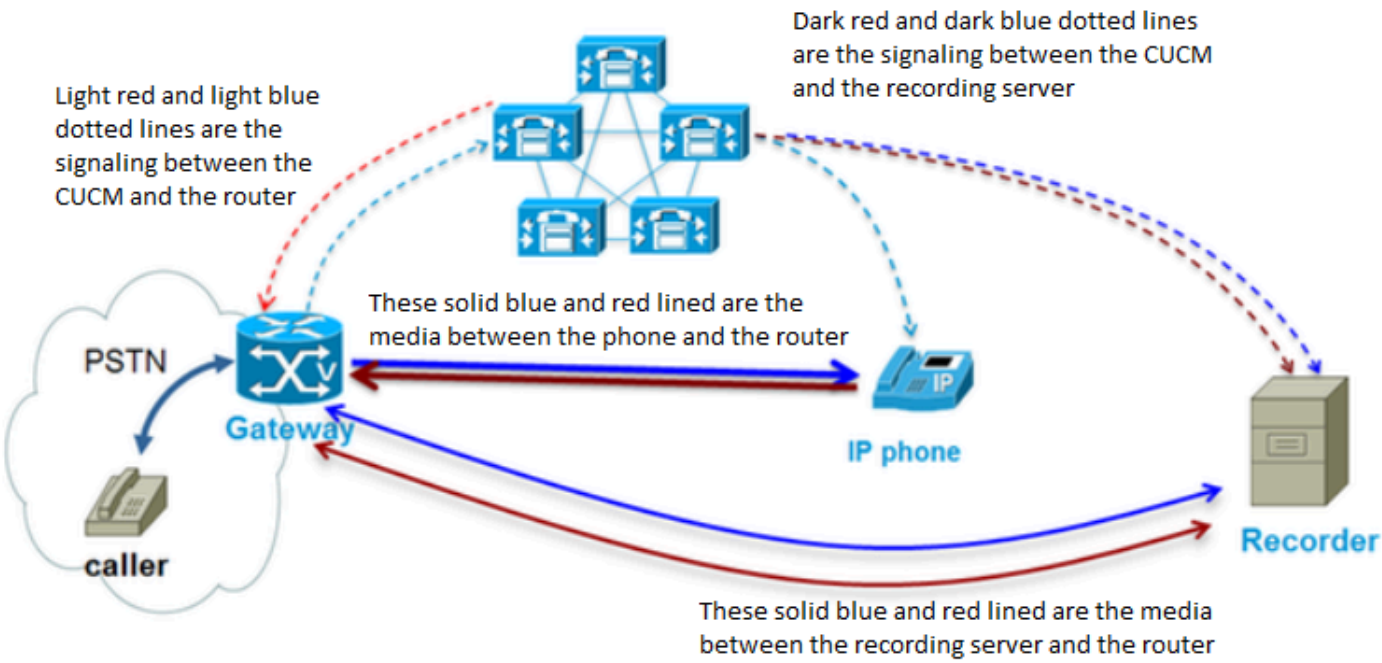



## Basé sur la passerelle

Les principaux éléments de l'enregistrement des appels basé sur la passerelle sont les suivants :

- La passerelle vocale achemine le support vers la destination d'enregistrement.
- CUCM s'enregistre auprès de la passerelle comme application.
- CUCM utilise le protocole HTTP afin de demander à Gateway (GW) de transmettre le contenu multimédia à la destination d'enregistrement.
- CUCM s'intègre à la destination d'enregistrement via une liaison SIP.
- Permet l'enregistrement des appels qui transitent simplement par le réseau géré (par exemple, pour les utilisateurs mobiles) ou pour les téléphones qui ne prennent pas en charge le BIB.

Comme vous pouvez le voir sur le schéma ici, le flux multimédia est très différent des autres types d'enregistrement d'appels :

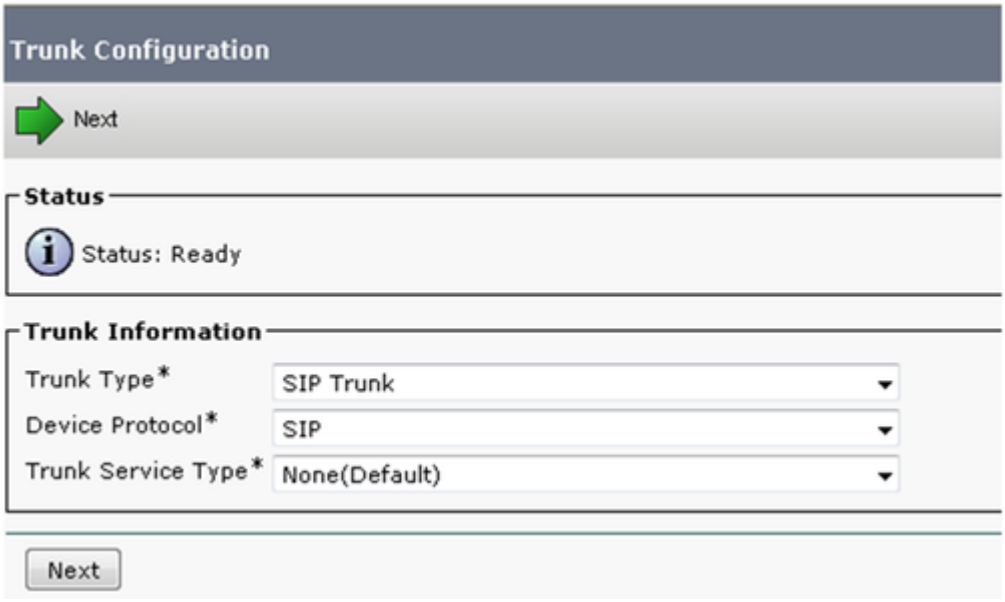



## Configuration de l'enregistrement automatique des appels pour l'intégration SIP uniquement

Cette section décrit comment configurer l'intégration SIP d'un serveur d'enregistrement.

### Créer une liaison SIP vers la destination d'enregistrement

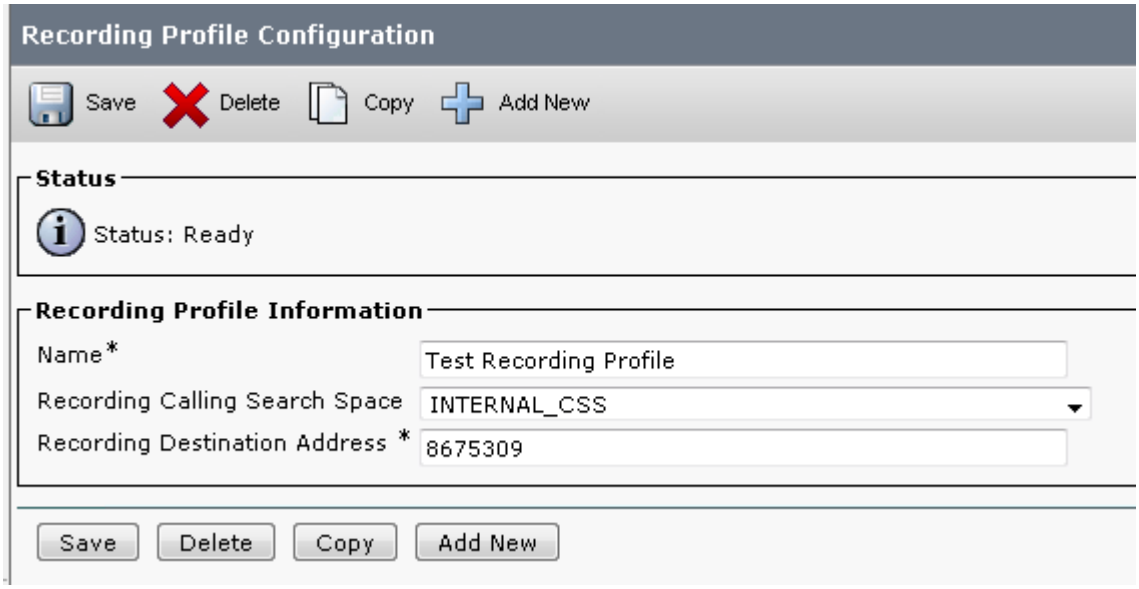
- Accédez à Device > Trunk, sélectionnez Add New.
- Créez une ligne principale SIP avec les paramètres indiqués dans l'image.



- Saisissez le nom du périphérique approprié, le pool de périphériques, MRGL, le profil de sécurité de la ligne principale SIP et le profil SIP.
- L'adresse de destination configurée est l'adresse du serveur d'application d'enregistrement.

## Créer un profil d'enregistrement

- Accédez à Device > Device Settings > Recording Profile.
- L'adresse de destination d'enregistrement est l'endroit où les appels d'enregistrement sont envoyés, comme illustré dans l'image.



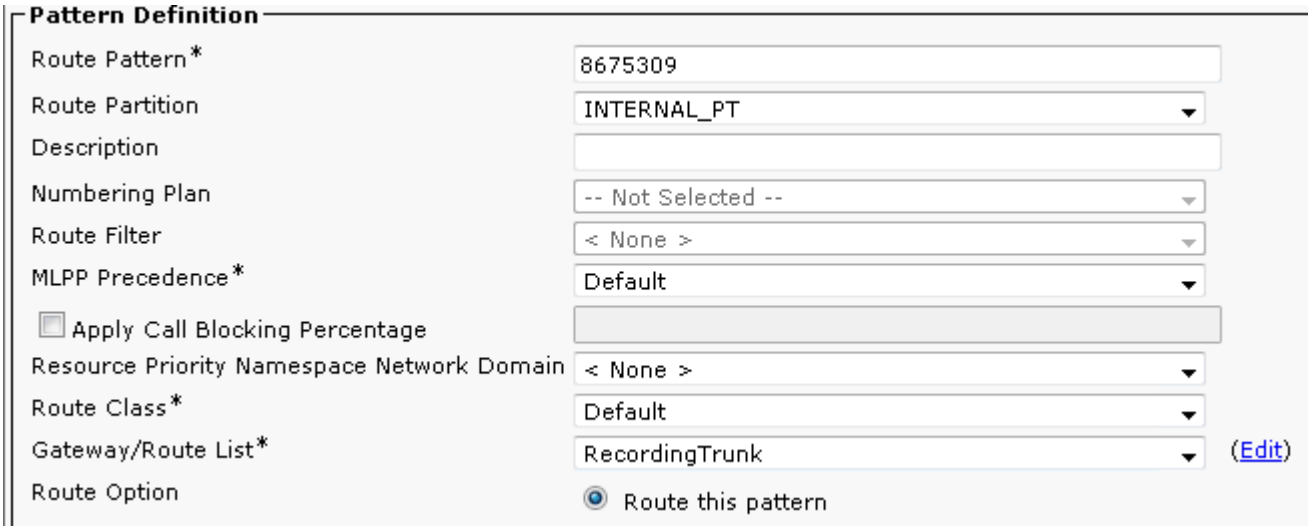

## Créer un modèle de routage pour router les appels d'enregistrement

- Créez un modèle de routage qui correspond à l'adresse de destination d'enregistrement configurée à l'étape précédente.
- Si vous souhaitez configurer des liaisons SIP redondantes, vous pouvez pointer vers une liste de routage au lieu de pointer directement vers la ligne principale SIP.

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Remarque : La partition attribuée à ce modèle de route doit être associée à l'espace RecordingCallingSearch et comme indiqué dans l'image.

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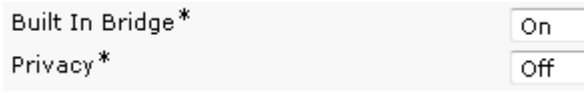

## Attribuer un profil d'enregistrement à une ligne téléphonique

- Sur un téléphone déjà créé avec un poste existant, affectez le profil d'enregistrement créé.
- Attribuez également le type d'enregistrement d'appel à cet emplacement.
- L'exemple montre l'enregistrement automatique, comme illustré dans l'image.




## Réglez BIB sur On et Confidentialité sur Off sur la page de configuration du téléphone

Dans la page de configuration du périphérique, accédez à la section intitulée Device Information. Réglez le pont intégré sur Activé et la confidentialité sur Désactivé, comme illustré dans l'image.




# Vérifier

Utilisez cette section pour confirmer que votre configuration fonctionne correctement.

Voici les comportements attendus dans les suivis Call Manager pour les téléphones SCCP et SIP

avec la configuration donnée. Ces exemples concernent un téléphone qui appelle un autre téléphone sur le même cluster alors que l'un des téléphones est configuré pour l'enregistrement des appels.

Remarque : Les journaux à collecter auprès de CUCM sont CTIManger, CallManager, Event Viewer App/Sys et des pcaps peuvent être nécessaires dans certains scénarios.

Remarque : Les journaux à collecter à partir des téléphones sont les journaux de console et pcaps. Vous pouvez obtenir des pcaps à partir du serveur d'enregistrement en même temps que vous obtenez les pcaps à partir du téléphone.

## SCCP

```
~~~~~~~~~~~~~~~~~~~~~~
Normal CCM Traces for SCCP phone to SCCP phone with SIP Integrated Call Recording
~~~~~~~~~~~~~~~~~~~~~~

### Calling phone places call

03796977.001 |20:21:08.055 |AppInfo  |StationInit: (0000109) SoftKeyEvent softKeyEvent=1(Redial) lineIn



### CUCM performs digit analysis against the dialed digits (dd="9110001")

03797017.001 |20:21:08.057 |AppInfo  |Digit Analysis: star_DaReq: daReq.partitionSearchSpace(), filtere
03797017.002 |20:21:08.057 |AppInfo  |Digit Analysis: star_DaReq: Matching Legacy Numeric, digits=91100
03797017.003 |20:21:08.057 |AppInfo  |Digit Analysis: getDaRes data&colon; daRes.ssType=[0] Intercept D
03797017.004 |20:21:08.057 |AppInfo  |Digit Analysis: getDaRes - Remote Destination [] isURI[1]
03797017.005 |20:21:08.057 |AppInfo  |Digit analysis: patternUsage=2
03797017.006 |20:21:08.057 |AppInfo  |Digit analysis: match(pi="2", fqcn="9110006", cn="9110006",plv="5
03797017.007 |20:21:08.057 |AppInfo  |Digit analysis: analysis results
03797017.008 |20:21:08.057 |AppInfo  ||PretransformCallingPartyNumber=9110006
|CallingPartyNumber=9110006
|DialingPartition=
|DialingPattern=9110001
|FullyQualifiedCalledPartyNumber=9110001
|DialingPatternRegularExpression=(9110001)
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(0,0,0)
|PretransformDigitString=9110001
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=9110001
|CollectedDigits=9110001



### CUCM determines call must stay on same node; go to LineControl (PID=LineControl(2,100,174,137))
```

```
03797019.001 |20:21:08.058 |AppInfo  |Digit analysis: wait_DmPidRes- Partition=[] Pattern=[9110001] Whe
```

### CUCM extends call to phone

```
03797036.003 |20:21:08.058 |AppInfo  |StationD:    (0000114) DEBUG whatToDo: line=1 calls=0 limit=4, bu
03797036.004 |20:21:08.058 |AppInfo  |StationD:    (0000114) DEBUG  whatToDo: busy trigger not hit... s
03797036.005 |20:21:08.058 |AppInfo  |preFilterCapCount =[11], preFilterCaps :: (Cap)= (25) (6) (4) (2)
03797036.006 |20:21:08.058 |AppInfo  |preFilterCapCount =[0], preFilterCaps :: (Cap)= Filtering Caps du
03797036.007 |20:21:08.058 |Created  |                                              |
03797036.008 |20:21:08.058 |AppInfo  |StationD:    (0000114) DEBUG- getLineRingSetting: retVal=4.
03797036.009 |20:21:08.058 |AppInfo  |StationD:    (0000114) DEBUG- saveRinger for: ci=38960750, line=1
03797036.010 |20:21:08.058 |AppInfo  |StationD:    (0000114) DEBUG- saveRinger: ci=38960750, line=1, mo
03797036.011 |20:21:08.058 |AppInfo  |StationD:    (0000114) INFO  sendCallAcceptReq: Try to send Stati
03797036.012 |20:21:08.058 |AppInfo  |StationD:    (0000114) playRinger for: ci=38960750.
03797036.013 |20:21:08.058 |AppInfo  |StationD:    (0000114) DEBUG- getLineRingSetting: retVal=4.
03797036.014 |20:21:08.058 |AppInfo  |StationD:    (0000114) DEBUG- getLineRingSetting: retVal=4.
03797036.015 |20:21:08.058 |AppInfo  |StationD:    (0000114) DEBUG- getLineRingSetting: retVal=4.
```

### Called (recorded) phone goes off hook

```
03797089.001 |20:21:09.335 |AppInfo  |StationD:    (0000114) restart0_StationOffHook - INFO: CI=3896075
```

### CUCM Tells the calling phone to open the logical channel

```
03797153.001 |20:21:09.337 |AppInfo  |StationD:    (0000109) SEP0018195AA209 , star_MediaExchangeAgenaO
```

### CUCM Tells the called (recorded party) phone to open the logical channel

```
03797156.001 |20:21:09.337 |AppInfo  |StationD:    (0000114) SEP001795BDD16B , star_MediaExchangeAgenaO
```

### CUCM Tells the calling phone to open the receive channel

```
03797164.002 |20:21:09.337 |AppInfo  |StationD:    (0000109) OpenReceiveChannel conferenceID=38960749 pa
```

### CUCM Tells the called (recorded party) phone to open the receive channel

```
03797168.002 |20:21:09.337 |AppInfo  |StationD:    (0000114) OpenReceiveChannel conferenceID=38960750 pa
```

### CUCM allocates BIB on called (recorded) phone

```
03797210.000 |20:21:09.338 |SdlSig   |MrmAllocateUcbResourceReq              |waiting



### BiB places first call to recording destination address (cn is calling party which is the BiB cn="b0

03797269.001 |20:21:09.340 |AppInfo  |Digit Analysis: star_DaReq: daReq.partitionSearchSpace(), filtere
03797269.002 |20:21:09.340 |AppInfo  |Digit Analysis: star_DaReq: Matching Legacy Numeric, digits=86753
03797269.003 |20:21:09.340 |AppInfo  |Digit Analysis: getDaRes data: daRes.ssType=[0] Intercept DAMR.ss
03797269.004 |20:21:09.340 |AppInfo  |Digit Analysis: getDaRes - Remote Destination [8675309] isURI[0]
03797269.005 |20:21:09.340 |AppInfo  |CMUtility routeCallThroughCTIRD: no matching RemDestDynamic recor
03797269.006 |20:21:09.340 |AppInfo  |DbMobility: getMatchedRemDest starts: cnumber = 8675309
03797269.007 |20:21:09.340 |AppInfo  |DbMobility: getMatchedRemDest: full match case
03797269.008 |20:21:09.340 |AppInfo  |DbMobility SelectByDestination: no matching RemDestDynamic record
03797269.009 |20:21:09.340 |AppInfo  |DbMobility: can't find remdest 8675309 in map
03797269.010 |20:21:09.340 |AppInfo  |Digit analysis: patternUsage=5
03797269.011 |20:21:09.340 |AppInfo  |Digit analysis: match(pi="1", fqcn="", cn="b00223908001",plv="5",
03797269.012 |20:21:09.340 |AppInfo  |Digit analysis: analysis results
03797269.013 |20:21:09.340 |AppInfo  ||PretransformCallingPartyNumber=b00223908001
|CallingPartyNumber=b00223908001
|DialingPartition=
|DialingPattern=8675309
|FullyQualifiedCalledPartyNumber=8675309
|DialingPatternRegularExpression=(8675309)
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(0,0,0)
|PretransformDigitString=8675309
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=8675309
|CollectedDigits=8675309



### CUCM sends INVITE #1 to configured recording server (10.48.32.170)

03797320.001 |20:21:09.343 |AppInfo  |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to 10.48
[212231,NET]
INVITE sip:8675309@10.48.32.170:5060 SIP/2.0
Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK204d520fedb3
From: <sip:9110001@10.48.32.90;x-nearend;x-refci=38960750;x-nearendclusterid=glenscucm10-5;x-nearenddev
To: <sip:8675309@10.48.32.170>
Date: Tue, 30 Sep 2014 00:21:09 GMT
Call-ID: abbb8e00-4291f775-204c-5a20300e@10.48.32.90
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM10.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: ;method="NOTIFY;Event=telephone-event;Duration=500"
Cisco-Guid: 2881195520-0000065536-0000000011-1512058894
Session-Expires:  1800
P-Asserted-Identity: <sip:9110001@10.48.32.90>
Remote-Party-ID: <sip:9110001@10.48.32.90>;party=calling;screen=yes;privacy=off
```

```
Contact: <sip:9110001@10.48.32.90:5060>;isFocus
Max-Forwards: 70
Content-Length: 0



### BiB places second call to recording destination address (cn is calling party which is the BiB cn="b
Note that the BiB number stayed the same (b00223908001) and so did the recordingdestination number

03797367.010 |20:21:09.344 |AppInfo  |Digit analysis: patternUsage=5
03797367.011 |20:21:09.344 |AppInfo  |Digit analysis: match(pi="1", fqcn="", cn="b00223908001",plv="5",
03797367.012 |20:21:09.344 |AppInfo  |Digit analysis: analysis results
03797367.013 |20:21:09.344 |AppInfo  ||PretransformCallingPartyNumber=b00223908001
|CallingPartyNumber=b00223908001
|DialingPartition=
|DialingPattern=8675309
|FullyQualifiedCalledPartyNumber=8675309
|DialingPatternRegularExpression=(8675309)
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(0,0,0)
|PretransformDigitString=8675309
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=8675309
|CollectedDigits=8675309



### CUCM receives 200 OK in response to INVITE #1

03797390.001 |20:21:09.345 |AppInfo  |//SIP/SIPUdp/wait_SdlDataInd: Incoming SIP UDP message size 737 f
[212232,NET]
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK204d520fedb3
From: <sip:9110001@10.48.32.90;x-nearend;x-refci=38960750;x-nearendclusterid=glenscucm10-5;x-nearenddev
To: <sip:8675309@10.48.32.170>;tag=1
Call-ID: abbb8e00-4291f775-204c-5a20300e@10.48.32.90
CSeq: 101 INVITE
Contact: <sip:10.48.32.170:5060;transport=udp>
Content-Type: application/sdp
Content-Length:   135

v=0
o=user1 53655765 2353687637 IN IP4 10.48.32.170
s=-
c=IN IP4 10.48.32.170
t=0 0
m=audio 6000 RTP/AVP 0
a=rtpmap:0 PCMU/8000



### CUCM sends INVITE #2 to recording server (10.48.32.170)

03797445.001 |20:21:09.348 |AppInfo  |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to 10.48
[212233,NET]
INVITE sip:8675309@10.48.32.170:5060 SIP/2.0
Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK204e754eaeae
```

```
From: <sip:9110001@10.48.32.90;x-farend;x-refci=3896075O;x-nearendclusterid=glenscucm10-5;x-nearenddevi
To: <sip:8675309@10.48.32.170>
Date: Tue, 30 Sep 2014 00:21:09 GMT
Call-ID: abbb8e00-4291f775-204d-5a20300e@10.48.32.90
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM10.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: ;method="NOTIFY;Event=telephone-event;Duration=500"
Cisco-Guid: 2881195520-0000065536-0000000012-1512058894
Session-Expires:  1800
P-Asserted-Identity: <sip:9110001@10.48.32.90>
Remote-Party-ID: <sip:9110001@10.48.32.90>;party=calling;screen=yes;privacy=off
Contact: <sip:9110001@10.48.32.90:5060>;isFocus
Max-Forwards: 70
Content-Length: 0
```

```
### CUCM receives 200 OK in response to INVITE #2

03797498.001 |20:21:09.350 |AppInfo  |//SIP/SIPUdp/wait_SdlDataInd: Incoming SIP UDP message size 736 f
[212235,NET]
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK204e754eaeae
From: <sip:9110001@10.48.32.90;x-farend;x-refci=3896075O;x-nearendclusterid=glenscucm10-5;x-nearenddevi
To: <sip:8675309@10.48.32.170>;tag=2
Call-ID: abbb8e00-4291f775-204d-5a20300e@10.48.32.90
CSeq: 101 INVITE
Contact: <sip:10.48.32.170:5060;transport=udp>
Content-Type: application/sdp
Content-Length:   135

v=0
o=user1 53655765 2353687637 IN IP4 10.48.32.170
s=-
c=IN IP4 10.48.32.170
t=0 0
m=audio 6000 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

```
### CUCM sends outbound ACK in response to 200 OK #1

03797500.001 |20:21:09.351 |AppInfo  |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to 10.48
[212236,NET]
ACK sip:10.48.32.170:5060;transport=UDP SIP/2.0
Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK204f50bef815
From: <sip:9110001@10.48.32.90;x-nearend;x-refci=3896075O;x-nearendclusterid=glenscucm10-5;x-nearenddev
To: <sip:8675309@10.48.32.170>;tag=1
Date: Tue, 30 Sep 2014 00:21:09 GMT
Call-ID: abbb8e00-4291f775-204c-5a20300e@10.48.32.90
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
```

```
CSeq: 101 ACK
Allow-Events: presence, kpml
Content-Type: application/sdp
Content-Length: 254

v=0
o=CiscoSystemsCCM-SIP 73601 1 IN IP4 10.48.32.90
s=SIP Call
c=IN IP4 10.48.32.33
b=TIAS:64000
b=CT:64
b=AS:64
t=0 0
m=audio 4000 RTP/AVP 0 101
a=ptime:20
a=rtpmap:0 PCMU/8000
a=sendonly
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15



### CUCM sends startMediaTransmission to the called (recorded) phone telling the phone to send RTP to r

03797479.001 |20:21:09.350 |AppInfo  |StationD:    (0000114) startMediaTransmission conferenceID=389607



### CUCM sends startMediaTransmission #2 to the called (recorded) phone telling the phone to send RTP t

03797596.001 |20:21:09.354 |AppInfo  |StationD:    (0000114) startMediaTransmission conferenceID=389607



### CUCM sends outbound ACK in response to 200 OK #2

03797615.001 |20:21:09.354 |AppInfo  |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to 10.48
[212237,NET]
ACK sip:10.48.32.170:5060;transport=UDP SIP/2.0
Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK2050183495f1
From: <sip:9110001@10.48.32.90;x-farend;x-refci=38960750;x-nearendclusterid=glenscucm10-5;x-nearenddevi
To: <sip:8675309@10.48.32.170>;tag=2
Date: Tue, 30 Sep 2014 00:21:09 GMT
Call-ID: abbb8e00-4291f775-204d-5a20300e@10.48.32.90
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Content-Type: application/sdp
Content-Length: 254

v=0
o=CiscoSystemsCCM-SIP 73602 1 IN IP4 10.48.32.90
s=SIP Call
c=IN IP4 10.48.32.33
b=TIAS:64000
b=CT:64
b=AS:64
t=0 0
```

```
m=audio 4000 RTP/AVP 0 101
a=ptime:20
a=rtpmap:0 PCMU/8000
a=sendonly
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15



### Calling phone sends CUCM the ORC ACK

03797634.001 |20:21:09.385 |AppInfo  |StationInit: (0000109) OpenReceiveChannelAck Status=0, IpAddr=IpA



### CUCM sends startMediaTransmission to the called (recorded) phone telling the phone to send RTP to t

03797642.001 |20:21:09.385 |AppInfo  |StationD:    (0000114) startMediaTransmission conferenceID=389607



### Called (recorded) phone sends CUCM the ORC ACK

03797643.001 |20:21:09.454 |AppInfo  |StationInit: (0000114) OpenReceiveChannelAck Status=0, IpAddr=IpA



### CUCM sends startMediaTransmission to the calling phone telling the phone to send RTP to the called

03797655.001 |20:21:09.454 |AppInfo  |StationD:    (0000109) startMediaTransmission conferenceID=
38960749 passThruPartyID=33554450 remoteIpAddress=IpAddr.type:0 ipAddr:0x0e3020210000000000000000000000
```

## SIP

```
~~~~~~~~~~~~~~~~~~~~~
Normal CCM Traces for SIP phone to SIP phone with SIP Integrated Call Recording
~~~~~~~~~~~~~~~~~~~~~

##### Calling phone places call


04241111.002 |11:27:41.232 |AppInfo  |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 10.48.38.
[286938,NET]
INVITE sip:1001@10.48.38.5;user=phone SIP/2.0
Via: SIP/2.0/TCP 10.48.38.102:50147;branch=z9hG4bK598c2eb2
From: "SJ User 1" <sip:1000@10.48.38.5>;tag=38ed18552a12296c00ff41e8-5fb7856e
To: <sip:1001@10.48.38.5>
Call-ID: 38ed1855-2a120006-78c34baf-1b81d864@10.48.38.102
Max-Forwards: 70
Session-ID: 1001532000105000a00038ed18552a12;remote=00000000000000000000000000000000
Date: Tue, 27 Aug 2019 15:27:42 GMT
CSeq: 101 INVITE
User-Agent: Cisco-CP7861/12.1.1
```

```
Contact: <sip:ab17ea6e-8072-927d-aad0-d10273906106@10.48.38.102:50147;transport=tcp>;+u.sip!devicename.
Expires: 180
Accept: application/sdp
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Remote-Party-ID: "SJ User 1" <sip:1000@10.48.38.5>;party=calling;id-type=subscriber;privacy=off;screen=
Supported: replaces,join,sdp-anat,norefersub,resource-priority,extended-refer,X-cisco-callinfo,X-cisco-
Allow-Events: kpml,dialog
Recv-Info: conference
Recv-Info: x-cisco-conference
Content-Length: 687
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 15384 0 IN IP4 10.48.38.102
s=SIP Call
b=AS:4064
t=0 0
m=audio 17904 RTP/AVP 114 9 113 115 0 8 116 18 101
c=IN IP4 10.48.38.102
b=TIAS:64000
a=rtpmap:114 opus/48000/2
a=fmtp:114 maxplaybackrate=16000;sprop-maxcapturerate=16000;maxaveragebitrate=64000;stereo=0;sprop-ster
a=rtpmap:9 G722/8000
a=rtpmap:113 AMR-WB/16000
a=fmtp:113 octet-align=0;mode-change-capability=2
a=rtpmap:115 AMR-WB/16000
a=fmtp:115 octet-align=1;mode-change-capability=2
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

##### CUCM performs digit analysis against the dialed digits (dd="1000")

```
04241138.007 |11:27:41.238 |AppInfo  |Digit analysis: match(pi="2", fqcn="+14085251000", cn="1000",plv=
04241138.008 |11:27:41.238 |AppInfo  |Digit analysis: analysis results
04241138.009 |11:27:41.238 |AppInfo  ||PretransformCallingPartyNumber=1000
|CallingPartyNumber=1000
|DialingPartition=INTERNAL_PT
|DialingPattern=1001
|FullyQualifiedCalledPartyNumber=+14085251001
|DialingPatternRegularExpression=(1001)
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(0,0,0)
|PretransformDigitString=1001
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=1001
|CollectedDigits=1001
```

##### CUCM determines call must stay on same node and go to LineControl (PID=LineControl(1,100,178,34)

04241140.001 |11:27:41.238 |AppInfo  |Digit analysis: wait_DmPidRes- Partition=[a067f454-fb26-2d1f-59da

##### CUCM sends outbound INVITE to called (recorded) phone

04241178.001 |11:27:41.242 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.
[286940,NET]
INVITE sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32e829c48246
From: "SJ User 1" <sip:1000@10.48.38.5>;tag=104952~e650e088-60ba-4195-8387-3dcc0127efdc-19301625
To: <sip:1001@10.48.38.5>
Date: Tue, 27 Aug 2019 15:27:41 GMT
Call-ID: 34241a00-d6514bed-327f-526300e@10.48.38.5
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM11.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= Unknown; orientation= from; gci= 1-2029; isVoip;
Send-Info: conference, x-cisco-conference
Alert-Info: <file://Bellcore-dr1/>
Session-ID: 1001532000105000a00038ed18552a12;remote=00000000000000000000000000000000
Remote-Party-ID: "SJ User 1" <sip:1000@10.48.38.5;x-cisco-callback-number=1000>;party=calling;screen=ye
Contact: <sip:1000@10.48.38.5:5060;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEP38ED18552A12"
Max-Forwards: 69
Content-Length: 0

##### Called (recorded) phone returns 200 OK

04241233.002 |11:27:43.614 |AppInfo  |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 10.48.38.
[286947,NET]
SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32e829c48246
From: "SJ User 1" <sip:1000@10.48.38.5>;tag=104952~e650e088-60ba-4195-8387-3dcc0127efdc-19301625
To: <sip:1001@10.48.38.5>;tag=6c416a369525006f33cf6f38-43c38ad2
Call-ID: 34241a00-d6514bed-327f-526300e@10.48.38.5
Session-ID: 4313758700105000a0006c416a369525;remote=1001532000105000a00038ed18552a12
Date: Tue, 27 Aug 2019 15:27:42 GMT
CSeq: 101 INVITE
Server: Cisco-CP7841/12.1.1
Contact: <sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp>;+u.sip!devicename.
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Remote-Party-ID: "SJ User 2" <sip:1001@10.48.38.5>;party=called;id-type=subscriber;privacy=off;screen=y
Supported: replaces,join,sdp-anat,norefersub,resource-priority,extended-refer,X-cisco-callinfo,X-cisco-
Allow-Events: kpml,dialog
Recv-Info: conference
Recv-Info: x-cisco-conference
Content-Length: 685
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 899 0 IN IP4 10.48.38.107

```
s=SIP Call
b=AS:4064
t=0 0
m=audio 20394 RTP/AVP 114 9 113 115 0 8 116 18 101
c=IN IP4 10.48.38.107
b=TIAS:64000
a=rtpmap:114 opus/48000/2
a=fmtp:114 maxplaybackrate=16000;sprop-maxcapturerate=16000;maxaveragebitrate=64000;stereo=0;sprop-ster
a=rtpmap:9 G722/8000
a=rtpmap:113 AMR-WB/16000
a=fmtp:113 octet-align=0;mode-change-capability=2
a=rtpmap:115 AMR-WB/16000
a=fmtp:115 octet-align=1;mode-change-capability=2
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:116 iLBC/8000
a=fmtp:116 mode=20
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=yes
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrec




### CUCM sends ACK to called (recorded) phone telling the called phone to send media to the calling pho

01314344.001 |11:18:48.652 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.48.32.
[106320,NET]
ACK sip:56ce4d7f-d3a2-40fd-a8b3-3f93c8832b9d@10.48.32.17:50841;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.48.32.90:5060;branch=z9hG4bK203c2831c118
From: <sip:9110006@10.48.32.90>;tag=38244~713e2333-4032-45f1-b1f5-e33cf471acec-47601638
To: <sip:9110011@10.48.32.90>;tag=b000b4d9e8cb0bba73e445ee-3cc7e650
Date: Tue, 14 Oct 2014 15:18:44 GMT
Call-ID: 6198e780-43d13ed4-203c-5a20300e@10.48.32.90
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence
Content-Type: application/sdp
Content-Length: 243

v=0
o=CiscoSystemsCCM-SIP 38244 1 IN IP4 10.48.32.90
s=SIP Call
c=IN IP4 10.48.32.28
b=TIAS:64000
b=CT:64
b=AS:64
t=0 0
m=audio 17260 RTP/AVP 0 101
a=ptime:20
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15### CUCM allocates BiB on called (recorded) phone

01314383.000 |11:18:48.675 |SdlSig   |MrmAllocateUcbResourceReq              |waiting
```

##### CUCM forwards the 200 OK to the calling phone

```
04241368.001 |11:27:43.624 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.
[286949,NET]
SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.48.38.102:50147;branch=z9hG4bK598c2eb2
From: "SJ User 1" <sip:1000@10.48.38.5>;tag=38ed18552a12296c00ff41e8-5fb7856e
To: <sip:1001@10.48.38.5>;tag=104951~e650e088-60ba-4195-8387-3dcc0127efdc-19301624
Date: Tue, 27 Aug 2019 15:27:41 GMT
Call-ID: 38ed1855-2a120006-78c34baf-1b81d864@10.48.38.102
CSeq: 101 INVITE
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
Allow-Events: presence
Supported: replaces
Server: Cisco-CUCM11.5
Call-Info: <urn:x-cisco-remotecc:callinfo>; security= NotAuthenticated; orientation= to; gci= 1-2029; i
Send-Info: conference, x-cisco-conference
Remote-Party-ID: "SJ User 2" <sip:1001@10.48.38.5>;party=called;screen=yes;privacy=off
Session-ID: 4313758700105000a0006c416a369525;remote=1001532000105000a00038ed18552a12
Remote-Party-ID: "SJ User 2" <sip:1001@10.48.38.5;user=phone>;party=x-cisco-original-called;privacy=off
Contact: <sip:1001@10.48.38.5:5060;transport=tcp>;+u.sip!devicename.ccm.cisco.com="SEP6C416A369525"
Content-Type: application/sdp
Content-Length: 223

v=0
o=CiscoSystemsCCM-SIP 104951 1 IN IP4 10.48.38.5
s=SIP Call
c=IN IP4 10.48.38.107
b=AS:64
t=0 0
m=audio 20394 RTP/AVP 0 101
b=TIAS:64000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

##### BiB allocation request on called (recorded) phone

```
04241393.000 |11:27:43.629 |SdlSig   |SIPAllocateBibResourceReq              |restart0
```

##### BiB allocated on called (recorded) phone

```
04241400.000 |11:27:43.630 |SdlSig   |MrmAllocateSharedResourceRes           |wait
```

##### DA for first call to activate BiB

```
04241418.006 |11:27:43.631 |AppInfo  |Digit analysis: match(pi="1", fqcn="", cn="",plv="5", pss="", Tod
04241418.007 |11:27:43.631 |AppInfo  |Digit analysis: analysis results
04241418.008 |11:27:43.631 |AppInfo  ||PretransformCallingPartyNumber=
|CallingPartyNumber=
|DialingPartition=
|DialingPattern=b0018615001
|FullyQualifiedCalledPartyNumber=b0018615001
```

|DialingPatternRegularExpression=(b0018615001)
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(1,86,15)
|PretransformDigitString=b0018615001
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=b0018615001
|CollectedDigits=b0018615001

##### CUCM sends INVITE #1 to called (recorded) phone with record-invoker=auto in Call-Info field and
Notice the SDP has a=inactive - even though there is no media established on the Bib yet.

04241449.001 |11:27:43.633 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.
[286950,NET]
INVITE sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32ea2a115cd6
From: "Call Manager" <sip:10.48.38.5>;tag=104956~e650e088-60ba-4195-8387-3dcc0127efdc-19301628
To: <sip:1001@10.48.38.5>
Date: Tue, 27 Aug 2019 15:27:43 GMT
Call-ID: 3555470O-d6514bef-3280-526300e@10.48.38.5
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM11.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence
Call-Info: <urn:x-cisco-remotecc:callinfo>; isVoip; record-invoker=auto
Join: 34241a00-d6514bed-327f-526300e@10.48.38.5;from-tag=6c416a369525006f33cf6f38-43c38ad2;to-tag=10495
Session-ID: 00000000000000000000000000000000;remote=00000000000000000000000000000000
Remote-Party-ID: "Call Manager" <sip:10.48.38.5>;party=calling;screen=yes;privacy=off
Contact: <sip:10.48.38.5:5060;transport=tcp>
Max-Forwards: 70
Content-Type: application/sdp
Content-Length: 187

v=0
o=CiscoSystemsCCM-SIP 104956 1 IN IP4 10.48.38.5
s=SIP Call
c=IN IP4 10.48.38.5
t=0 0
m=audio 4000 RTP/AVP 0
a=label:X-relay-nearend
a=rtpmap:0 PCMU/8000
a=inactive
a=mid:1

##### Calling phone sends CUCM an ACK in response to the 200 OK which was from when the user at the ca

04241455.002 |11:27:43.697 |AppInfo  |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 10.48.38.
[286951,NET]
ACK sip:1001@10.48.38.5:5060;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.48.38.102:50147;branch=z9hG4bK688db3c1
From: "SJ User 1" <sip:1000@10.48.38.5>;tag=38ed18552a12296c00ff41e8-5fb7856e
To: <sip:1001@10.48.38.5>;tag=104951~e650e088-60ba-4195-8387-3dcc0127efdc-19301624

```
Call-ID: 38ed1855-2a120006-78c34baf-1b81d864@10.48.38.102
Max-Forwards: 70
Session-ID: 1001532000105000a00038ed18552a12;remote=4313758700105000a0006c416a369525
Date: Tue, 27 Aug 2019 15:27:45 GMT
CSeq: 101 ACK
User-Agent: Cisco-CP7861/12.1.1
Remote-Party-ID: "SJ User 1" <sip:1000@10.48.38.5>;party=calling;id-type=subscriber;privacy=off;screen=
Content-Length: 0
Recv-Info: conference
Recv-Info: x-cisco-conference
```

##### Called (recorded) phone returns 200 OK in response to the invite with "record-invoker=auto"
Notice the SDP has a=inactive - even though there is no media established on the Bib yet.

```
04241466.002 |11:27:43.901 |AppInfo  |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 10.48.38.
[286953,NET]
SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32ea2a115cd6
From: "Call Manager" <sip:10.48.38.5>;tag=104956~e650e088-60ba-4195-8387-3dcc0127efdc-19301628
To: <sip:1001@10.48.38.5>;tag=6c416a369525007019bf48f9-5901eb85
Call-ID: 35554700-d6514bef-3280-526300e@10.48.38.5
Session-ID: 0848153900105000a0006c416a369525;remote=00000000000000000000000000000000
Date: Tue, 27 Aug 2019 15:27:42 GMT
CSeq: 101 INVITE
Server: Cisco-CP7841/12.1.1
Contact: <sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp>;+u.sip!devicename.
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Remote-Party-ID: "SJ User 2" <sip:1001@10.48.38.5>;party=called;id-type=subscriber;privacy=off;screen=y
Supported: replaces,join,sdp-anat,norefersub,resource-priority,extended-refer,X-cisco-callinfo,X-cisco-
Allow-Events: kpml,dialog
Recv-Info: conference
Recv-Info: x-cisco-conference
Content-Length: 218
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 2684 0 IN IP4 10.48.38.107
s=SIP Call
t=0 0
m=audio 26396 RTP/AVP 0 101
c=IN IP4 10.48.38.107
b=TIAS:64000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=inactive
```

##### CUCM responds to called (recorded) phone with ACK

```
04241469.001 |11:27:43.901 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.
[286954,NET]
ACK sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32eb34decb69
From: "Call Manager" <sip:10.48.38.5>;tag=104956~e650e088-60ba-4195-8387-3dcc0127efdc-19301628
To: <sip:1001@10.48.38.5>;tag=6c416a369525007019bf48f9-5901eb85
```

```
Date: Tue, 27 Aug 2019 15:27:43 GMT
Call-ID: 35554700-d6514bef-3280-526300e@10.48.38.5
User-Agent: Cisco-CUCM11.5
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence
Content-Length: 0
```

##### BiB places first call to recording destination address (cn is calling party which is the BiB cn=

```
04241501.011 |11:27:43.905 |AppInfo  |Digit analysis: match(pi="1", fqcn="", cn="b0018615001",plv="5", |
04241501.012 |11:27:43.905 |AppInfo  |Digit analysis: analysis results
04241501.013 |11:27:43.905 |AppInfo  ||PretransformCallingPartyNumber=b0018615001
|CallingPartyNumber=b0018615001
|DialingPartition=INTERNAL_PT
|DialingPattern=7878
|FullyQualifiedCalledPartyNumber=7878
|DialingPatternRegularExpression=(7878)
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(0,0,0)
|PretransformDigitString=7878
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=7878
|CollectedDigits=7878
```

##### DA for to activate BiB for the other person's side of the call

```
04241545.006 |11:27:43.907 |AppInfo  |Digit analysis: match(pi="1", fqcn="", cn="",plv="5", pss="", Tod
04241545.007 |11:27:43.907 |AppInfo  |Digit analysis: analysis results
04241545.008 |11:27:43.907 |AppInfo  ||PretransformCallingPartyNumber=
|CallingPartyNumber=
|DialingPartition=
|DialingPattern=b0018615001
|FullyQualifiedCalledPartyNumber=b0018615001
|DialingPatternRegularExpression=(b0018615001)
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(1,86,15)
|PretransformDigitString=b0018615001
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=b0018615001
|CollectedDigits=b0018615001
```

##### CUCM sends INVITE #1 to configured recording server (10.48.38.30)

```
04241555.001 |11:27:43.908 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.
[286955,NET]
INVITE sip:7878@10.48.38.30:5060 SIP/2.0
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32ecc2c802c
From: "SJ User 2" <sip:1001@10.48.38.5;x-nearend;x-refci=19301625;x-nearendclusterid=StandAloneCluster;
```

```
To: <sip:7878@10.48.38.30>
Date: Tue, 27 Aug 2019 15:27:43 GMT
Call-ID: 35554700-d6514bef-3281-526300e@10.48.38.5
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM11.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: <sip:10.48.38.5:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Call-Info: <urn:x-cisco-remotecc:callinfo>;x-cisco-video-traffic-class=DESKTOP
Session-ID: 0848153900105000a0006c416a369525;remote=00000000000000000000000000000000
Cisco-Guid: 0894781184-0000065536-0000000022-0086388750
Session-Expires:  1800
P-Asserted-Identity: "SJ User 2" <sip:1001@10.48.38.5>
Remote-Party-ID: "SJ User 2" <sip:1001@10.48.38.5>;party=calling;screen=yes;privacy=off
Contact: <sip:1001@10.48.38.5:5060;transport=tcp>;isFocus;+u.sip!devicename.ccm.cisco.com="SEP6C416A369
Max-Forwards: 70
Content-Length: 0
```

##### CUCM sends INVITE #2 to called (recorded) phone with record-invoker=auto in Call-Info field and
Notice the SDP has a=inactive - even though there is no media established on the Bib yet.

```
04241590.001 |11:27:43.910 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.
[286956,NET]
INVITE sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32ed62f39668
From: "Call Manager" <sip:10.48.38.5>;tag=104959~e650e088-60ba-4195-8387-3dcc0127efdc-19301631
To: <sip:1001@10.48.38.5>
Date: Tue, 27 Aug 2019 15:27:43 GMT
Call-ID: 35554700-d6514bef-3282-526300e@10.48.38.5
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM11.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence
Call-Info: <urn:x-cisco-remotecc:callinfo>; isVoip; record-invoker=auto
Join: 34241a00-d6514bed-327f-526300e@10.48.38.5;from-tag=6c416a369525006f33cf6f38-43c38ad2;to-tag=10495
Session-ID: 00000000000000000000000000000000;remote=00000000000000000000000000000000
Remote-Party-ID: "Call Manager" <sip:10.48.38.5>;party=calling;screen=yes;privacy=off
Contact: <sip:10.48.38.5:5060;transport=tcp>
Max-Forwards: 70
Content-Type: application/sdp
Content-Length: 186

v=0
o=CiscoSystemsCCM-SIP 104959 1 IN IP4 10.48.38.5
s=SIP Call
c=IN IP4 10.48.38.5
t=0 0
m=audio 4000 RTP/AVP 0
a=label:X-relay-farend
a=rtpmap:0 PCMU/8000
a=inactive
```

```
a=mid:1
```

##### Called (recorded) phone returns 200 OK in response to INVITE #2 to invoke BiB
Notice the SDP has a=inactive - even though there is no media established on the Bib yet.

```
04241614.002 |11:27:44.197 |AppInfo  |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 10.48.38.
[286959,NET]
SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32ed62f39668
From: "Call Manager" <sip:10.48.38.5>;tag=104959~e650e088-60ba-4195-8387-3dcc0127efdc-19301631
To: <sip:1001@10.48.38.5>;tag=6c416a369525007145d433c8-062b13d7
Call-ID: 3555470O-d6514bef-3282-526300e@10.48.38.5
Session-ID: 56a8a95e00105000a0006c416a369525;remote=00000000000000000000000000000000
Date: Tue, 27 Aug 2019 15:27:42 GMT
CSeq: 101 INVITE
Server: Cisco-CP7841/12.1.1
Contact: <sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp>;+u.sip!devicename.
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Remote-Party-ID: "SJ User 2" <sip:1001@10.48.38.5>;party=called;id-type=subscriber;privacy=off;screen=y
Supported: replaces,join,sdp-anat,norefersub,resource-priority,extended-refer,X-cisco-callinfo,X-cisco-
Allow-Events: kpml,dialog
Recv-Info: conference
Recv-Info: x-cisco-conference
Content-Length: 219
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 13977 0 IN IP4 10.48.38.107
s=SIP Call
t=0 0
m=audio 17904 RTP/AVP 0 101
c=IN IP4 10.48.38.107
b=TIAS:64000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=inactive
```

##### CUCM responds with ACK for 200 OK for INVITE #2 to invoke the BiB

```
04241618.001 |11:27:44.199 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.
[286960,NET]
ACK sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32ee41b380b1
From: "Call Manager" <sip:10.48.38.5>;tag=104959~e650e088-60ba-4195-8387-3dcc0127efdc-19301631
To: <sip:1001@10.48.38.5>;tag=6c416a369525007145d433c8-062b13d7
Date: Tue, 27 Aug 2019 15:27:43 GMT
Call-ID: 3555470O-d6514bef-3282-526300e@10.48.38.5
User-Agent: Cisco-CUCM11.5
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence
Content-Length: 0
```

##### BiB places second call to recording destination address (cn is calling party which is the BiB cn

```
04241651.011 |11:27:44.201 |AppInfo  |Digit analysis: match(pi="1", fqcn="", cn="b0018615001",plv="5", |
04241651.012 |11:27:44.202 |AppInfo  |Digit analysis: analysis results
04241651.013 |11:27:44.202 |AppInfo  ||PretransformCallingPartyNumber=b0018615001
|CallingPartyNumber=b0018615001
|DialingPartition=INTERNAL_PT
|DialingPattern=7878
|FullyQualifiedCalledPartyNumber=7878
|DialingPatternRegularExpression=(7878)
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(0,0,0)
|PretransformDigitString=7878
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=7878
|CollectedDigits=7878
```

##### CUCM sends INVITE #2 to configured recording server

```
04241698.001 |11:27:44.205 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.
[286961,NET]
INVITE sip:7878@10.48.38.30:5060 SIP/2.0
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32ef2867938b
From: "SJ User 2" <sip:1001@10.48.38.5;x-farend;x-refci=19301625;x-nearendclusterid=StandAloneCluster;x
To: <sip:7878@10.48.38.30>
Date: Tue, 27 Aug 2019 15:27:44 GMT
Call-ID: 35eddd80-d6514bf0-3283-526300e@10.48.38.5
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM11.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: <sip:10.48.38.5:5060>;method="NOTIFY;Event=telephone-event;Duration=500"
Call-Info: <urn:x-cisco-remotecc:callinfo>;x-cisco-video-traffic-class=DESKTOP
Session-ID: 56a8a95e00105000a0006c416a369525;remote=00000000000000000000000000000000
Cisco-Guid: 0904781184-0000065536-0000000023-0086388750
Session-Expires:  1800
P-Asserted-Identity: "SJ User 2" <sip:1001@10.48.38.5>
Remote-Party-ID: "SJ User 2" <sip:1001@10.48.38.5>;party=calling;screen=yes;privacy=off
Contact: <sip:1001@10.48.38.5:5060;transport=tcp>;isFocus;+u.sip!devicename.ccm.cisco.com="SEP6C416A369
Max-Forwards: 70
Content-Length: 0
```

##### CUCM receives a 200 OK from recording server for INVITE #2

```
04241723.002 |11:27:44.324 |AppInfo  |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 10.48.38.
[286963,NET]
SIP/2.0 200 Ok
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32ef2867938b
```

```
To: <sip:7878@10.48.38.30>;tag=ds1a1d776c
From: "SJ User 2" <sip:1001@10.48.38.5;x-farend;x-refci=19301625;x-nearendclusterid=StandAloneCluster;x
Call-ID: 35eddd80-d6514bf0-3283-526300e@10.48.38.5
CSeq: 101 INVITE
Content-Length: 475
Contact: <sip:7878@10.48.38.30:5060;transport=TCP>
Content-Type: application/sdp
Allow: INVITE, BYE, CANCEL, ACK, NOTIFY, INFO, UPDATE
Supported: X-cisco-srtp-fallback
Server: MediaSense/11.x

v=0
o=CiscoORA 707 1 IN IP4 10.48.38.30
s=SIP Call
c=IN IP4 10.48.38.30
t=0 0
m=audio 56512 RTP/SAVP 102 0 8 9 18
a=rtpmap:102 MP4A-LATM/90000
a=fmtp:102 profile-level-id=24;object=23;bitrate=64000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:9 G722/8000
a=rtpmap:18 G729/8000
a=recvonly
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
```

##### CUCM receives 200 OK from the recording server in response to INVITE #1

```
04241743.002 |11:27:44.326 |AppInfo  |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 10.48.38.
[286964,NET]
SIP/2.0 200 Ok
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32ecc2c802c
To: <sip:7878@10.48.38.30>;tag=ds2c967644
From: "SJ User 2" <sip:1001@10.48.38.5;x-nearend;x-refci=19301625;x-nearendclusterid=StandAloneCluster;
Call-ID: 3555470-d6514bef-3281-526300e@10.48.38.5
CSeq: 101 INVITE
Content-Length: 475
Contact: <sip:7878@10.48.38.30:5060;transport=TCP>
Content-Type: application/sdp
Allow: INVITE, BYE, CANCEL, ACK, NOTIFY, INFO, UPDATE
Supported: X-cisco-srtp-fallback
Server: MediaSense/11.x

v=0
o=CiscoORA 708 1 IN IP4 10.48.38.30
s=SIP Call
c=IN IP4 10.48.38.30
t=0 0
m=audio 59058 RTP/SAVP 102 0 8 9 18
a=rtpmap:102 MP4A-LATM/90000
a=fmtp:102 profile-level-id=24;object=23;bitrate=64000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:9 G722/8000
a=rtpmap:18 G729/8000
a=recvonly
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
a=crypto:XXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXXX
```

##### CUCM sends re-INVITE #2 to called (recorded) phone (notice there is no SDP - this is so CUCM can
Notice there is no SDP

```
04241825.001 |11:27:44.330 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.
[286965,NET]
INVITE sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32f014677161
From: "Call Manager" <sip:10.48.38.5>;tag=104959~e650e088-60ba-4195-8387-3dcc0127efdc-19301631
To: <sip:1001@10.48.38.5>;tag=6c416a369525007145d433c8-062b13d7
Date: Tue, 27 Aug 2019 15:27:44 GMT
Call-ID: 3555470O-d6514bef-3282-526300e@10.48.38.5
Supported: timer,resource-priority,replaces
User-Agent: Cisco-CUCM11.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, SUBSCRIBE, NOTIFY
CSeq: 102 INVITE
Max-Forwards: 70
Expires: 180
Allow-Events: presence
Call-Info: <urn:x-cisco-remotecc:callinfo>; isVoip; record-invoker=auto
Min-SE:  1800
Session-ID: 00000000000000000000000000000000;remote=56a8a95e00105000a0006c416a369525
Remote-Party-ID: "Call Manager" <sip:10.48.38.5>;party=calling;screen=yes;privacy=off
Contact: <sip:10.48.38.5:5060;transport=tcp>
Content-Length: 0
```

##### CUCM sends re-INVITE #1 to called (recorded) phone (notice there is no SDP - this is so CUCM can

```
04241866.001 |11:27:44.332 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.
[286966,NET]
INVITE sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32f11da4ce39
From: "Call Manager" <sip:10.48.38.5>;tag=104956~e650e088-60ba-4195-8387-3dcc0127efdc-19301628
To: <sip:1001@10.48.38.5>;tag=6c416a369525007019bf48f9-5901eb85
Date: Tue, 27 Aug 2019 15:27:44 GMT
Call-ID: 3555470O-d6514bef-3280-526300e@10.48.38.5
Supported: timer,resource-priority,replaces
User-Agent: Cisco-CUCM11.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, SUBSCRIBE, NOTIFY
CSeq: 102 INVITE
Max-Forwards: 70
Expires: 180
Allow-Events: presence
Call-Info: <urn:x-cisco-remotecc:callinfo>; isVoip; record-invoker=auto
Min-SE:  1800
Session-ID: 00000000000000000000000000000000;remote=0848153900105000a0006c416a369525
Remote-Party-ID: "Call Manager" <sip:10.48.38.5>;party=calling;screen=yes;privacy=off
Contact: <sip:10.48.38.5:5060;transport=tcp>
Content-Length: 0
```

##### Called (recorded) phone returns 200 OK for re-INVITE #2

```
04241872.002 |11:27:44.541 |AppInfo  |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 10.48.38.
[286969,NET]
```

```
SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32f014677161
From: "Call Manager" <sip:10.48.38.5>;tag=104959~e650e088-60ba-4195-8387-3dcc0127efdc-19301631
To: <sip:1001@10.48.38.5>;tag=6c416a369525007145d433c8-062b13d7
Call-ID: 3555470O-d6514bef-3282-526300e@10.48.38.5
Session-ID: 56a8a95e00105000a0006c416a369525;remote=00000000000000000000000000000000
Date: Tue, 27 Aug 2019 15:27:43 GMT
CSeq: 102 INVITE
Server: Cisco-CP7841/12.1.1
Contact: <sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp>;+u.sip!devicename.
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Remote-Party-ID: "SJ User 2" <sip:1001@10.48.38.5>;party=called;id-type=subscriber;privacy=off;screen=y
Supported: replaces,join,sdp-anat,norefersub,resource-priority,extended-refer,X-cisco-callinfo,X-cisco-
Allow-Events: kpml,dialog
Recv-Info: conference
Recv-Info: x-cisco-conference
Content-Length: 219
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 13977 1 IN IP4 10.48.38.107
s=SIP Call
t=0 0
m=audio 17904 RTP/AVP 0 101
c=IN IP4 10.48.38.107
b=TIAS:64000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

##### Called (recorded) phone returns 200 OK to re-INVITE #1

```
04241885.002 |11:27:44.550 |AppInfo  |SIPTcp - wait_SdlReadRsp: Incoming SIP TCP message from 10.48.38.
[286970,NET]
SIP/2.0 200 OK
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32f11da4ce39
From: "Call Manager" <sip:10.48.38.5>;tag=104956~e650e088-60ba-4195-8387-3dcc0127efdc-19301628
To: <sip:1001@10.48.38.5>;tag=6c416a369525007019bf48f9-5901eb85
Call-ID: 3555470O-d6514bef-328O-526300e@10.48.38.5
Session-ID: 0848153900105000a0006c416a369525;remote=00000000000000000000000000000000
Date: Tue, 27 Aug 2019 15:27:43 GMT
CSeq: 102 INVITE
Server: Cisco-CP7841/12.1.1
Contact: <sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp>;+u.sip!devicename.
Allow: ACK,BYE,CANCEL,INVITE,NOTIFY,OPTIONS,REFER,REGISTER,UPDATE,SUBSCRIBE,INFO
Remote-Party-ID: "SJ User 2" <sip:1001@10.48.38.5>;party=called;id-type=subscriber;privacy=off;screen=y
Supported: replaces,join,sdp-anat,norefersub,resource-priority,extended-refer,X-cisco-callinfo,X-cisco-
Allow-Events: kpml,dialog
Recv-Info: conference
Recv-Info: x-cisco-conference
Content-Length: 218
Content-Type: application/sdp
Content-Disposition: session;handling=optional

v=0
o=Cisco-SIPUA 2684 1 IN IP4 10.48.38.107
s=SIP Call
```

```
t=0 0
m=audio 26396 RTP/AVP 0 101
c=IN IP4 10.48.38.107
b=TIAS:64000
a=rtpmap:0 PCMU/8000
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=sendrecv
```

##### CUCM sends ACK to called (recorded) phone for re-INVITE #2

```
04241903.001 |11:27:44.552 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.
[286971,NET]
ACK sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32f252b587f6
From: "Call Manager" <sip:10.48.38.5>;tag=104959~e650e088-60ba-4195-8387-3dcc0127efdc-19301631
To: <sip:1001@10.48.38.5>;tag=6c416a369525007145d433c8-062b13d7
Date: Tue, 27 Aug 2019 15:27:44 GMT
Call-ID: 35554700-d6514bef-3282-526300e@10.48.38.5
User-Agent: Cisco-CUCM11.5
Max-Forwards: 70
CSeq: 102 ACK
Allow-Events: presence
Session-ID: 00000000000000000000000000000000;remote=56a8a95e00105000a0006c416a369525
Content-Type: application/sdp
Content-Length: 192

v=0
o=CiscoSystemsCCM-SIP 104959 3 IN IP4 10.48.38.5
s=SIP Call
c=IN IP4 10.48.38.30
b=TIAS:64000
b=AS:64
t=0 0
m=audio 56512 RTP/AVP 0
b=TIAS:64000
a=rtpmap:0 PCMU/8000
a=recvonly
```

##### CUCM sends ACK to the recording server in response to 200 OK #2

```
04241917.001 |11:27:44.555 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.
[286972,NET]
ACK sip:7878@10.48.38.30:5060;transport=TCP SIP/2.0
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32f373e69393
From: "SJ User 2" <sip:1001@10.48.38.5;x-farend;x-refci=19301625;x-nearendclusterid=StandAloneCluster;x
To: <sip:7878@10.48.38.30>;tag=ds1a1d776c
Date: Tue, 27 Aug 2019 15:27:44 GMT
Call-ID: 35eddd80-d6514bf0-3283-526300e@10.48.38.5
User-Agent: Cisco-CUCM11.5
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Session-ID: 56a8a95e00105000a0006c416a369525;remote=c83405810147c69016c38634ab104961
Content-Type: application/sdp
Content-Length: 235
```

```
v=0
o=CiscoSystemsCCM-SIP 104961 1 IN IP4 10.48.38.5
s=SIP Call
c=IN IP4 10.48.38.107
b=TIAS:64000
b=AS:64
t=0 0
m=audio 17904 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=sendonly
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

##### CUCM sends ACK to called (recorded) phone for re-INVITE #1

```
04241947.001 |11:27:44.559 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.
[286973,NET]
ACK sip:91a43f66-ca58-9cd3-b0e5-588aa61a72bc@10.48.38.107:51902;transport=tcp SIP/2.0
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32f45d25b711
From: "Call Manager" <sip:10.48.38.5>;tag=104956~e650e088-60ba-4195-8387-3dcc0127efdc-19301628
To: <sip:1001@10.48.38.5>;tag=6c416a369525007019bf48f9-5901eb85
Date: Tue, 27 Aug 2019 15:27:44 GMT
Call-ID: 35554700-d6514bef-3280-526300e@10.48.38.5
User-Agent: Cisco-CUCM11.5
Max-Forwards: 70
CSeq: 102 ACK
Allow-Events: presence
Session-ID: 00000000000000000000000000000000;remote=0848153900105000a0006c416a369525
Content-Type: application/sdp
Content-Length: 192

v=0
o=CiscoSystemsCCM-SIP 104956 3 IN IP4 10.48.38.5
s=SIP Call
c=IN IP4 10.48.38.30
b=TIAS:64000
b=AS:64
t=0 0
m=audio 59058 RTP/AVP 0
b=TIAS:64000
a=rtpmap:0 PCMU/8000
a=recvonly
```

##### CUCM sends ACK to the recording server in response to 200 OK #1

```
04241948.001 |11:27:44.559 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP message to 10.48.38.
[286974,NET]
ACK sip:7878@10.48.38.30:5060;transport=TCP SIP/2.0
Via: SIP/2.0/TCP 10.48.38.5:5060;branch=z9hG4bK32f573871bbb
From: "SJ User 2" <sip:1001@10.48.38.5;x-nearend;x-refci=19301625;x-nearendclusterid=StandAloneCluster;
To: <sip:7878@10.48.38.30>;tag=ds2c967644
Date: Tue, 27 Aug 2019 15:27:43 GMT
Call-ID: 35554700-d6514bef-3281-526300e@10.48.38.5
User-Agent: Cisco-CUCM11.5
Max-Forwards: 70
```

```
CSeq: 101 ACK
Allow-Events: presence, kpml
Session-ID: 0848153900105000a0006c416a369525;remote=c83405810147c69016c38634ab104958
Content-Type: application/sdp
Content-Length: 235

v=0
o=CiscoSystemsCCM-SIP 104958 1 IN IP4 10.48.38.5
s=SIP Call
c=IN IP4 10.48.38.107
b=TIAS:64000
b=AS:64
t=0 0
m=audio 26396 RTP/AVP 0 101
a=rtpmap:0 PCMU/8000
a=sendonly
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

# Dépannage

Cette section fournit des informations que vous pouvez utiliser pour dépanner votre configuration.

## Négociation du codec

Voici un exemple de l'un des types les plus courants d'échecs d'enregistrement d'appels : non-concordance de codec entre le téléphone enregistré et le serveur d'enregistrement :

```
~~~~~~~~~~~~~~~~~~~~~~~~~
Codec Negotiation Failure
~~~~~~~~~~~~~~~~~~~~~~~~~

### Calling phone places call

00019629.001 |12:48:34.510 |AppInfo  |StationInit: (0000005) EnblocCall calledParty=9110001.



### CUCM performs digit analysis against the dialed digits (dd="9110001")

00019638.001 |12:48:34.511 |AppInfo  |Digit Analysis: star_DaReq: daReq.partitionSearchSpace(), filtere
00019638.002 |12:48:34.511 |AppInfo  |Digit Analysis: star_DaReq: Matching Legacy Numeric, digits=91100
00019638.003 |12:48:34.522 |AppInfo  |Digit Analysis: getDaRes data: daRes.ssType=[0] Intercept DAMR.ss
00019638.004 |12:48:34.522 |AppInfo  |Digit Analysis: getDaRes - Remote Destination [] isURI[1]
00019638.005 |12:48:34.522 |AppInfo  |Digit analysis: patternUsage=2
00019638.006 |12:48:34.522 |AppInfo  |Digit analysis: match(pi="2", fqcn="9110006", cn="9110006",plv="5
00019638.007 |12:48:34.522 |AppInfo  |Digit analysis: analysis results
```

```
00019638.008 |12:48:34.522 |AppInfo  ||PretransformCallingPartyNumber=9110006
|CallingPartyNumber=9110006
|DialingPartition=
|DialingPattern=9110001
|FullyQualifiedCalledPartyNumber=9110001
|DialingPatternRegularExpression=(9110001)
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(0,0,0)
|PretransformDigitString=9110001
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=9110001
|CollectedDigits=9110001
```

```
### CUCM determines call must stay on same node and go to LineControl (PID=LineControl(2,100,174,19))

00019640.001 |12:48:34.522 |AppInfo  |Digit analysis: wait_DmPidRes- Partition=[] Pattern=[9110001] Whe
```

```
### CUCM extends the call to the called phone

00019657.003 |12:48:34.560 |AppInfo  |StationD:    (0000007) DEBUG whatToDo: line=1 calls=0 limit=4, bu
00019657.004 |12:48:34.560 |AppInfo  |StationD:    (0000007) DEBUG  whatToDo: busy trigger not hit... s
00019657.005 |12:48:34.560 |AppInfo  |preFilterCapCount =[11], preFilterCaps :: (Cap)= (25) (6) (4) (2)
00019657.006 |12:48:34.560 |AppInfo  |preFilterCapCount =[0], preFilterCaps :: (Cap)= Filtering Caps du
00019657.007 |12:48:34.560 |Created  |                                          |
00019657.008 |12:48:34.560 |AppInfo  |StationD:    (0000007) DEBUG- getLineRingSetting: retVal=4.
00019657.009 |12:48:34.560 |AppInfo  |StationD:    (0000007) DEBUG- saveRinger for: ci=49613638, line=1
00019657.010 |12:48:34.560 |AppInfo  |StationD:    (0000007) DEBUG- saveRinger: ci=49613638, line=1, mo
00019657.011 |12:48:34.560 |AppInfo  |StationD:    (0000007) INFO  sendCallAcceptReq: Try to send Stati
00019657.012 |12:48:34.560 |AppInfo  |StationD:    (0000007) playRinger for: ci=49613638.
00019657.013 |12:48:34.560 |AppInfo  |StationD:    (0000007) DEBUG- getLineRingSetting: retVal=4.
00019657.014 |12:48:34.560 |AppInfo  |StationD:    (0000007) DEBUG- getLineRingSetting: retVal=4.
00019657.015 |12:48:34.560 |AppInfo  |StationD:    (0000007) DEBUG- getLineRingSetting: retVal=4.
```

```
### The Called (recorded) phone goes off hook

00019709.001 |12:48:36.042 |AppInfo  |StationD:    (0000007) restart0_StationOffHook - INFO: CI=4961363
```

```
### CUCM Tells the calling phone to open the logical channel

00019773.001 |12:48:36.061 |AppInfo  |StationD:    (0000005) SEP0018195AA209 , star_MediaExchangeAgenaO
```

```
### CUCM Tells the called (recorded) to open the logical channel

00019776.001 |12:48:36.061 |AppInfo  |StationD:    (0000007) SEP001795BDD16B , star_MediaExchangeAgenaO
```

### CUCM Tells the calling phone to open the receive channel

00019784.002 |12:48:36.062 |AppInfo  |StationD:    (0000005) OpenReceiveChannel conferenceID=49613637 pa

### Codec locked due to recording on called (recorded) phone

00019785.003 |12:48:36.062 |AppInfo  | StationCdpc: star_MediaExchangeAgenaQueryCapability - Device SEP

### CUCM Tells the called (recorded) phone to open the receive channel

00019788.002 |12:48:36.062 |AppInfo  |StationD:    (0000007) OpenReceiveChannel conferenceID=49613638 pa

### CUCM allocates the BiB on the called (recorded) phone

00019830.000 |12:48:36.074 |SdlSig   |MrmAllocateUcbResourceReq              |waiting

### BiB places it's first call to recording destination address (cn is calling number which is the BiB

00019889.001 |12:48:36.100 |AppInfo  |Digit Analysis: star_DaReq: daReq.partitionSearchSpace(), filtere
00019889.002 |12:48:36.100 |AppInfo  |Digit Analysis: star_DaReq: Matching Legacy Numeric, digits=86753
00019889.003 |12:48:36.100 |AppInfo  |Digit Analysis: getDaRes data: daRes.ssType=[0] Intercept DAMR.ss
00019889.004 |12:48:36.100 |AppInfo  |Digit Analysis: getDaRes - Remote Destination [8675309] isURI[0]
00019889.005 |12:48:36.100 |AppInfo  |CMUtility routeCallThroughCTIRD: no matching RemDestDynamic recor
00019889.006 |12:48:36.100 |AppInfo  |DbMobility: getMatchedRemDest starts: cnumber = 8675309
00019889.007 |12:48:36.100 |AppInfo  |DbMobility: getMatchedRemDest: full match case
00019889.008 |12:48:36.100 |AppInfo  |DbMobility SelectByDestination: no matching RemDestDynamic record
00019889.009 |12:48:36.100 |AppInfo  |DbMobility: can't find remdest 8675309 in map
00019889.010 |12:48:36.100 |AppInfo  |Digit analysis: patternUsage=5
00019889.011 |12:48:36.100 |AppInfo  |Digit analysis: match(pi="1", fqcn="", cn="b00223906001",plv="5",
00019889.012 |12:48:36.100 |AppInfo  |Digit analysis: analysis results
00019889.013 |12:48:36.100 |AppInfo  ||PretransformCallingPartyNumber=b00223906001
|CallingPartyNumber=b00223906001
|DialingPartition=
|DialingPattern=8675309
|FullyQualifiedCalledPartyNumber=8675309
|DialingPatternRegularExpression=(8675309)
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(0,0,0)
|PretransformDigitString=8675309
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=8675309
|CollectedDigits=8675309

### Calling phone sends CUCM the ORC ACK

00019912.001 |12:48:36.139 |AppInfo  |StationInit: (0000005) OpenReceiveChannelAck Status=0, IpAddr=IpA

### CUCM sends startMediaTransmission to the called (recorded) phone telling the phone to send RTP to t

00019920.001 |12:48:36.139 |AppInfo  |StationD:    (0000007) startMediaTransmission conferenceID=496136

### Called (recorded) phone sends CUCM the ORC ACK

00019959.001 |12:48:36.145 |AppInfo  |StationInit: (0000007) OpenReceiveChannelAck Status=0, IpAddr=IpA

### CUCM sends startMediaTransmission to the calling phone telling the phone to send RTP to the called

00019977.001 |12:48:36.146 |AppInfo  |StationD:    (0000005) startMediaTransmission conferenceID=496136

### BiB places second call to recording destination address (cn is calling number which is the BiB cn="
Note that the BiB number stayed the same (b00223906001) and so did the recordingdestination number

00020002.001 |12:48:36.147 |AppInfo  |Digit Analysis: star_DaReq: daReq.partitionSearchSpace(), filtere
00020002.002 |12:48:36.147 |AppInfo  |Digit Analysis: star_DaReq: Matching Legacy Numeric, digits=86753
00020002.003 |12:48:36.147 |AppInfo  |Digit Analysis: getDaRes data: daRes.ssType=[0] Intercept DAMR.ss
00020002.004 |12:48:36.147 |AppInfo  |Digit Analysis: getDaRes - Remote Destination [8675309] isURI[0]
00020002.005 |12:48:36.147 |AppInfo  |CMUtility routeCallThroughCTIRD: no matching RemDestDynamic recor
00020002.006 |12:48:36.147 |AppInfo  |DbMobility: getMatchedRemDest starts: cnumber = 8675309
00020002.007 |12:48:36.147 |AppInfo  |DbMobility: getMatchedRemDest: full match case
00020002.008 |12:48:36.147 |AppInfo  |DbMobility SelectByDestination: no matching RemDestDynamic record
00020002.009 |12:48:36.147 |AppInfo  |DbMobility: can't find remdest 8675309 in map
00020002.010 |12:48:36.147 |AppInfo  |Digit analysis: patternUsage=5
00020002.011 |12:48:36.147 |AppInfo  |Digit analysis: match(pi="1", fqcn="", cn="b00223906001",plv="5",
00020002.012 |12:48:36.147 |AppInfo  |Digit analysis: analysis results
00020002.013 |12:48:36.147 |AppInfo  ||PretransformCallingPartyNumber=b00223906001
|CallingPartyNumber=b00223906001
|DialingPartition=
|DialingPattern=8675309
|FullyQualifiedCalledPartyNumber=8675309
|DialingPatternRegularExpression=(8675309)
|DialingWhere=
|PatternType=Enterprise
|PotentialMatches=NoPotentialMatchesExist
|DialingSdlProcessId=(0,0,0)
|PretransformDigitString=8675309
|PretransformTagsList=SUBSCRIBER
|PretransformPositionalMatchList=8675309
|CollectedDigits=8675309
|UnconsumedDigits=
|TagsList=SUBSCRIBER
|PositionalMatchList=8675309

### CUCM sends INVITE #1 to configured recording server (10.48.32.170)

```
00020086.001 |12:48:36.156 |AppInfo  |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to 10.48
[901,NET]
INVITE sip:8675309@10.48.32.170:5060 SIP/2.0
Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK4f2a857d3d
From: <sip:9110001@10.48.32.90;x-nearend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-nearenddev
To: <sip:8675309@10.48.32.170>
Date: Tue, 14 Oct 2014 16:48:36 GMT
Call-ID: ef7acf80-43d153e4-50-5a20300e@10.48.32.90
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM10.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: ;method="NOTIFY;Event=telephone-event;Duration=500"
Cisco-Guid: 4017803136-0000065536-0000000001-1512058894
Session-Expires:  1800
P-Asserted-Identity: <sip:9110001@10.48.32.90>
Remote-Party-ID: <sip:9110001@10.48.32.90>;party=calling;screen=yes;privacy=off
Contact: <sip:9110001@10.48.32.90:5060>;isFocus
Max-Forwards: 70
Content-Length: 0
```

### CUCM sends INVITE #2 to configured recording server (10.48.32.170)

```
00020088.001 |12:48:36.157 |AppInfo  |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to 10.48
[902,NET]
INVITE sip:8675309@10.48.32.170:5060 SIP/2.0
Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK5014378d0b
From: <sip:9110001@10.48.32.90;x-farend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-nearenddevi
To: <sip:8675309@10.48.32.170>
Date: Tue, 14 Oct 2014 16:48:36 GMT
Call-ID: ef7acf80-43d153e4-51-5a20300e@10.48.32.90
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM10.5
Allow: INVITE, OPTIONS, INFO, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY
CSeq: 101 INVITE
Expires: 180
Allow-Events: presence, kpml
Supported: X-cisco-srtp-fallback
Supported: Geolocation
Call-Info: ;method="NOTIFY;Event=telephone-event;Duration=500"
Cisco-Guid: 4017803136-0000065536-0000000002-1512058894
Session-Expires:  1800
P-Asserted-Identity: <sip:9110001@10.48.32.90>
Remote-Party-ID: <sip:9110001@10.48.32.90>;party=calling;screen=yes;privacy=off
Contact: <sip:9110001@10.48.32.90:5060>;isFocus
Max-Forwards: 70
Content-Length: 0
```

```
### CUCM receives a 200 OK from recording server for INVITE #1

00020089.001 |12:48:36.161 |AppInfo  |//SIP/SIPUdp/wait_SdlDataInd: Incoming SIP UDP message size 731 f
[903,NET]
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK4f2a857d3d
From: <sip:9110001@10.48.32.90;x-nearend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-nearenddev
To: <sip:8675309@10.48.32.170>;tag=1
Call-ID: ef7acf80-43d153e4-50-5a20300e@10.48.32.90
CSeq: 101 INVITE
Contact: <sip:10.48.32.170:5060;transport=udp>
Content-Type: application/sdp
Content-Length:   135

v=0
o=user1 53655765 2353687637 IN IP4 10.48.32.170
s=-
c=IN IP4 10.48.32.170
t=0 0
m=audio 6000 RTP/AVP 0
a=rtpmap:0 PCMU/8000




### CUCM receives a 200 OK from recording server for INVITE #2

00020092.001 |12:48:36.161 |AppInfo  |//SIP/SIPUdp/wait_SdlDataInd: Incoming SIP UDP message size 730 f
[905,NET]
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK5014378d0b
From: <sip:9110001@10.48.32.90;x-farend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-nearenddevi
To: <sip:8675309@10.48.32.170>;tag=2
Call-ID: ef7acf80-43d153e4-51-5a20300e@10.48.32.90
CSeq: 101 INVITE
Contact: <sip:10.48.32.170:5060;transport=udp>
Content-Type: application/sdp
Content-Length:   135

v=0
o=user1 53655765 2353687637 IN IP4 10.48.32.170
s=-
c=IN IP4 10.48.32.170
t=0 0
m=audio 6000 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

<#root>

```
### Region information for connecting audio for recording call, both appear to support G.711.
Note that the bandwidth capabilities printed is kbps=8 meaning the region relationship between the two

00020160.005 |12:48:36.190 |AppInfo  |DET-RegionsServer::matchCapabilities-- savedOption=3, PREF_NONE,
00020160.006 |12:48:36.190 |AppInfo  |DET-MediaManager-(2)::checkAudioPassThru, param(bPostMTPAllocatio
00020160.007 |12:48:36.190 |AppInfo  |DET-MediaManager-(2)::preCheckCapabilities,
```

**region1=Default**

,

**region2=RecordingTrunk**

,

**Pty1**

capCount=1 (Cap,ptime)=

**(4,20)**

,

**Pty2**

capCount=1 (Cap,ptime)=

**(4,20)**

00020160.008 |12:48:36.190 |AppInfo |DET-RegionsServer::matchCapabilities-- savedOption=0, PREF_NONE,

**kbps=8**

, capACount=1, capBCount=1

<#root>

### CUCM determines 2 transcoders are required and attempts to allocate

00020160.011 |12:48:36.190 |AppInfo |DET-MediaManager-(2)::preCheckCapabilities,

**caps mismatch! Xcoder Reqd. kbps(8)**

, filtered A[capCount=0 (Cap,ptime)=], B[capCount=0 (Cap,ptime)=] allowMTP=0

**numXcoderRequired=2**

xcodingSide=0

### No transcoder is configured which can cause this call to fail

00020162.003 |12:48:36.190 |AppInfo |MediaResourceManager::sendAllocationResourceErr - ERROR - no tran

<#root>

### CUCM sendt the ACK and BYE to the recording server in response to INVITE #1
Note the Q.850 cause code

```
00020210.001 |12:48:36.216 |AppInfo  |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to 10.48
[906,NET]
ACK sip:10.48.32.170:5060;transport=UDP SIP/2.0
Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK51257b2b47
From: <sip:9110001@10.48.32.90;x-nearend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-nearenddev
To: <sip:8675309@10.48.32.170>;tag=1
Date: Tue, 14 Oct 2014 16:48:36 GMT
Call-ID: ef7acf80-43d153e4-50-5a20300e@10.48.32.90
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Content-Length: 0


00020211.001 |12:48:36.216 |AppInfo |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to 10.48.
[907,NET]
BYE sip:10.48.32.170:5060;transport=UDP SIP/2.0
Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK526f3d2afa
From: <sip:9110001@10.48.32.90;x-nearend;x-refci=49613638;x-nearendclusterid=GlensCUCM10-5;x-nearenddev
To: <sip:8675309@10.48.32.170>;tag=1
Date: Tue, 14 Oct 2014 16:48:36 GMT
Call-ID: ef7acf80-43d153e4-50-5a20300e@10.48.32.90
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
P-Asserted-Identity: <sip:9110001@10.48.32.90>
CSeq: 102 BYE

Reason: Q.850;cause=47


Content-Length: 0
```

<#root>

```
### CUCM sendt the ACK and BYE to the recording server in response to INVITE #2
Note the Q.850 cuase code in the BYE

00020248.001 |12:48:36.218 |AppInfo  |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to 10.48
[908,NET]
ACK sip:10.48.32.170:5060;transport=UDP SIP/2.0
Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK531df920a6
From: <sip:9110001@10.48.32.90;x-farend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-nearenddevi
To: <sip:8675309@10.48.32.170>;tag=2
Date: Tue, 14 Oct 2014 16:48:36 GMT
Call-ID: ef7acf80-43d153e4-51-5a20300e@10.48.32.90
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: presence, kpml
Content-Length: 0


00020249.001 |12:48:36.218 |AppInfo  |//SIP/SIPUdp/wait_SdlSPISignal: Outgoing SIP UDP message to 10.48
[909,NET]
BYE sip:10.48.32.170:5060;transport=UDP SIP/2.0
Via: SIP/2.0/UDP 10.48.32.90:5060;branch=z9hG4bK5462aba807
From: <sip:9110001@10.48.32.90;x-farend;x-refci=49613638;x-nearendclusterid=glenscucm10-5;x-nearenddevi
To: <sip:8675309@10.48.32.170>;tag=2
```

```
Date: Tue, 14 Oct 2014 16:48:36 GMT
Call-ID: ef7acf80-43d153e4-51-5a20300e@10.48.32.90
User-Agent: Cisco-CUCM10.5
Max-Forwards: 70
P-Asserted-Identity: <sip:9110001@10.48.32.90>
CSeq: 102 BYE

Reason: Q.850;cause=47

Content-Length: 0
```

## Mauvaise configuration incluant des problèmes CSS et PT

Les commandes disponibles ici permettent de vérifier rapidement la plupart des configurations d'enregistrement avec uniquement l'adresse MAC connue d'un téléphone qui n'enregistre pas d'appels. Remplacez simplement la partie de la commande MAC_of_Phone par l'adresse MAC réelle du téléphone, comme dans les exemples présentés ici.

Vous obtenez ainsi le DN (tous s'il y en a plusieurs) de l'adresse MAC sur laquelle vous effectuez la recherche, l'adresse MAC du téléphone uniquement pour confirmation, le paramètre BIB, le paramètre de confidentialité, le type d'enregistrement (référencez les valeurs répertoriées dans les exemples des travaux pratiques), le profil d'enregistrement utilisé par le téléphone, le nom des espaces de recherche d'appels (CSS) d'enregistrement, la destination d'enregistrement de ce profil d'enregistrement et la partition à laquelle cette destination d'enregistrement est associée en fonction de l'adresse MAC sur laquelle vous effectuez la recherche :

```
run sql select n1.dnorpattern as phone_dn, dev.name as phone_mac, CASE dev.tkstatus_builtinbridge WHEN
```

Vous obtenez ainsi la liste des partitions associées à la CSS d'enregistrement sur le profil d'enregistrement associé à l'adresse MAC du téléphone sur lequel vous effectuez la recherche.

```
run sql select css.name as name_of_the_recording_css, rp.name as partitions_in_recording_css, csm.sorto
```

Voici des exemples de résultats de travaux pratiques pour un téléphone dont l'adresse MAC est SEPC80084AA8743 :

Dans cette commande, vous pouvez voir que le téléphone n'a qu'un seul DN sur lui qui est 2003, nous voyons également le BIB est activé, la confidentialité est désactivée, le type d'enregistrement est automatique, la source préférée est le téléphone, le profil d'enregistrement est Test Recording Profile, l'espace de recherche d'appel d'enregistrement est INTERNAL_CSS, le modèle de route

pour les appels enregistrés est 8675309 et ce modèle est associé à la partition INTERNAL_PT.

```
run sql select n1.dnorpattern as phone_dn, dev.name as phone_mac, CASE dev.tkstatus_builtinbridge WHEN
phone_dn phone_mac       is_bib_on  is_privacy_on  recording_type     recording_media_source recording_
======== =============== ========== ============== ================== ====================== ==========
2003     SEPC80084AA8743 BiB is on  Privacy is off Automatic          Phone Preferred        Test Recor
```

Avec le résultat de cette commande, vous pouvez vérifier toutes les partitions de l'enregistrement CSS et du profil d'enregistrement associé au téléphone d'intérêt. Vous pouvez voir ici que la partition INTERNAL_PT est l'une des partitions associées à l'espace de recherche d'appels INTERNAL_CSS. Cela signifie qu'il ne doit y avoir aucun problème avec le BIB du téléphone qui est en mesure d'appeler le modèle de route d'enregistrement.

<#root>

```
run sql select css.name as name_of_the_recording_css, rp.name as partitions_in_recording_css, csm.sorto
```

**SEPC80084AA8743**

```
'
name_of_the_recording_css partitions_in_recording_css sortorder
========================= ============================ =========
INTERNAL_CSS              E911_PT                      1
INTERNAL_CSS              Phones_PT                    2
INTERNAL_CSS              EMERGENCY_PT                 3
```

**INTERNAL_CSS** **INTERNAL_PT**

```
                4
INTERNAL_CSS              INFORMACAST_PT               5
```

# Informations connexes

- Cisco Collaboration System 11.x Solution Reference Network Designs (SRND)

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